Review & Exploration of Audio Watermarking using Empirical Mode of Decomposition

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ABSTRACT

In this paper we deal with the review of Audio files are capable of hiding the data , The identity of the owner of the audio file can be hidden in the audio file which is called Watermark. Watermarking is the process of embedding information into a signal (e.g. audio, video or pictures) in a way that is difficult to remove. If the signal is copied, then the information is also carried in the copy. An audio watermark is a unique electronic identifier embedded in an audio signal, typically used to identify ownership of copyright. Audio watermarking is done using Empirical Mode Decomposition (EMD) where the audio signal is divided into segments and then decomposed into different IMFs (Intrinsic Mode Functions).The watermark is then applied to the last IMF as binary bits using QIM (Quantization Index Modulation).

Keyword: *Empirical mode decomposition, intrinsic mode function, audio watermarking, image,DCT*

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1. INTRODUCTION

Watermarking is the method of hiding information bits into a host signal. The information bits or data can be used for identifying the owner, for copyright protection, authentication, copy control etc. The host signal can be Audio, video or image. The copyright protection of digital media makes their cities smarter and intelligent in order to take a decision at real-time based on current city scenarios. The complete system description is depicted in next sections.

Digital audio watermarking has received a great deal of attention in the literature to provide efficient solutions for copyright protection of digital media by embedding a watermark in the original audio signal. Main requirements of digital audio watermarking are imperceptibility, robustness and data capacity. More precisely, the watermark must be inaudible within the host audio data to maintain audio quality and robust to signal distortions applied to the host data. Finally, the watermark must be easy to extract to prove ownership. There are different methods for audio watermarking. The methods are broadly classified into temporal and spectral watermarking. In temporal watermarking the watermarked data are embedded directly into the host audio signal in the time domain. In spectral watermarking certain frequency transforms are performed to the host audio signal and then embed the watermark info into the transformed frequency domain data block. In temporal watermarking many techniques are adopted such as watermarking in the dual channel audio using echo hiding scheme [1], echo hiding using analysis by synthesis method [2], using time scale modification method [3], using EMD (Empirical Mode Decomposition) method [4].In spectral watermarking many techniques are used which involves the transformation of the host signal

2. Literature Review

Wei FOO et.al. [7] gave an adaptive algorithm for audio watermarking which uses echo hiding method. The algorithm has been divided into two parts encoder design and decoder design. In the encoder section, segmentation is performed on the original audio signal. After which mask computation and echo hiding is done and watermark information along with the kernel parameters are embedded into the audio signal. The segments are processed and watermark checking is performed to obtain the data about the watermark position and finally after recombining the segments, watermark audio signal is obtained. In the decoder design part of the algorithm, first the audio segmentation is performed on the watermarked audio signal. If two channels are obtained then peal detection is performed on the left and right channels and the respective results are compared and if consistent bits are obtained then the bits are combined and then watermark is used to identify the owner. Various attacks were made to check the robustness of the technique used and it has been concluded that the robustness against filtering is less.

3. REQUIREMENTS OF AUDIO WATERMARK

A.Capacity: Watermarking capacity always depends on the amount of data that can be embedded into a host signal. Generally sound signal, capacity requirement always depend on imperceptibility and hardiness. Higher capacity is usually obtained at the expense of either hardiness or imperceptibility.

B.Imperceptibility: Most importantly, the watermark signal should be imperceptible to the end user who is listening to or viewing the host signal. This means that a typical user should not be able to differentiate between watermarked and un watermarked signals. The watermark signal should be in cognizable because the presence or absence of a watermark should not subtract from the first purpose of the host signal, that of transfer high-quality audio or visual data. Additionally, perceptible distortion might indicate the presence of a watermark, and maybe its precise location within a host signal. This data information is also used by a malicious party to distort, replace, or take away the watermark information.

C. Asymmetry: If for the complete set of cover objects the watermark remains same; then, extracting for one file can cause harm watermark of all the files. Thus, imbalance is also an understandable concern. It's suggested to possess distinctive watermarks to completely different files to help build the technique more helpful.

D. Robustness: Watermarks must be able to handle all sorts of attacks whether it is conversion, compression, Noise or any other form of attack.

E. Speed: The speed of embedding of watermark is very important in real time applications wherever the embedding is completed on continuous signals like, speech of a politician or language between pilot and communication system workers. a number of the attainable applications wherever speed may be a constraint are audio streaming and airline traffic observation. Each embedding and extraction method ought to be created as quick as attainable with greater efficiency.
the remote medical treatment and care, call-

4. PROPOSED METHOD OF AUDIO WATERMARKING VIA EMD

EMD is fully data-driven method that recursively breaks down any signal into a reduced number of zero-mean with symmetric envelopes AM-FM components called Intrinsic Mode Functions (IMFs). The audio signal is divided into frames and each one is decomposed adaptively, by EMD, into intrinsic oscillatory components called Intrinsic Mode Functions (IMFs). The watermark and the synchronization codes are embedded into the extreme of the last IMF, a low frequency mode stable under different attacks and preserving audio perceptual quality of the host signal. [1]

Watermark Embedding:-

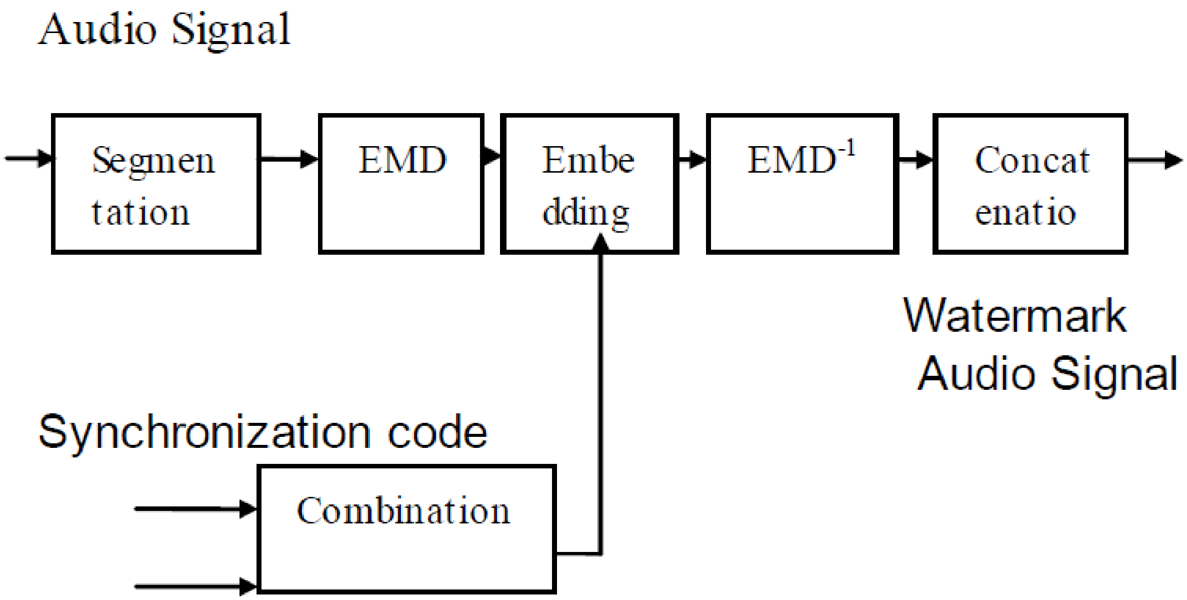


Fig 4.1 Watermark Embedding

Watermark Extraction:-

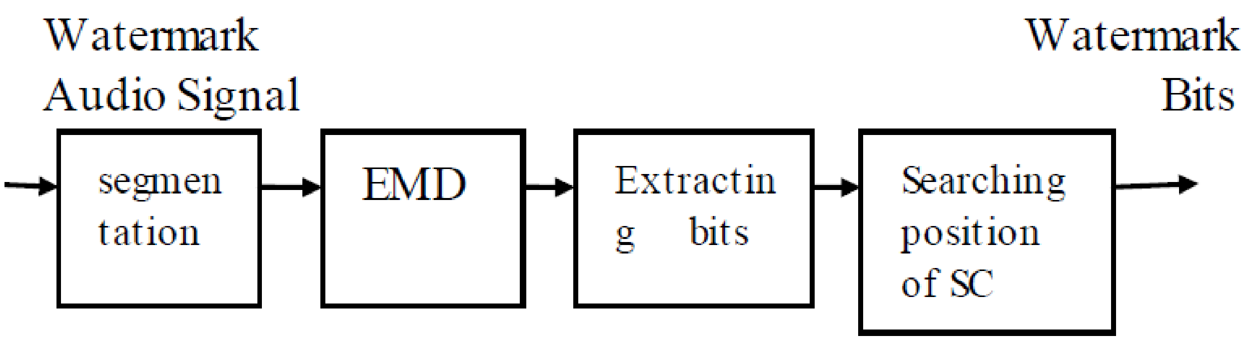


Fig 4.2 Watermark Extraction

5. PERFORMANCE ANALYSIS

We evaluate the performance of our method in terms of data payload, error probability of SC, Signal to Noise Ratio (SNR) between original and the watermarked audio signals, Bit Error Rate and Normalized cross-Correlation . According to International Federation of the Photographic Industry (IFPI) recommendations, a watermark audio signal should maintain more than 20 dB SNR.

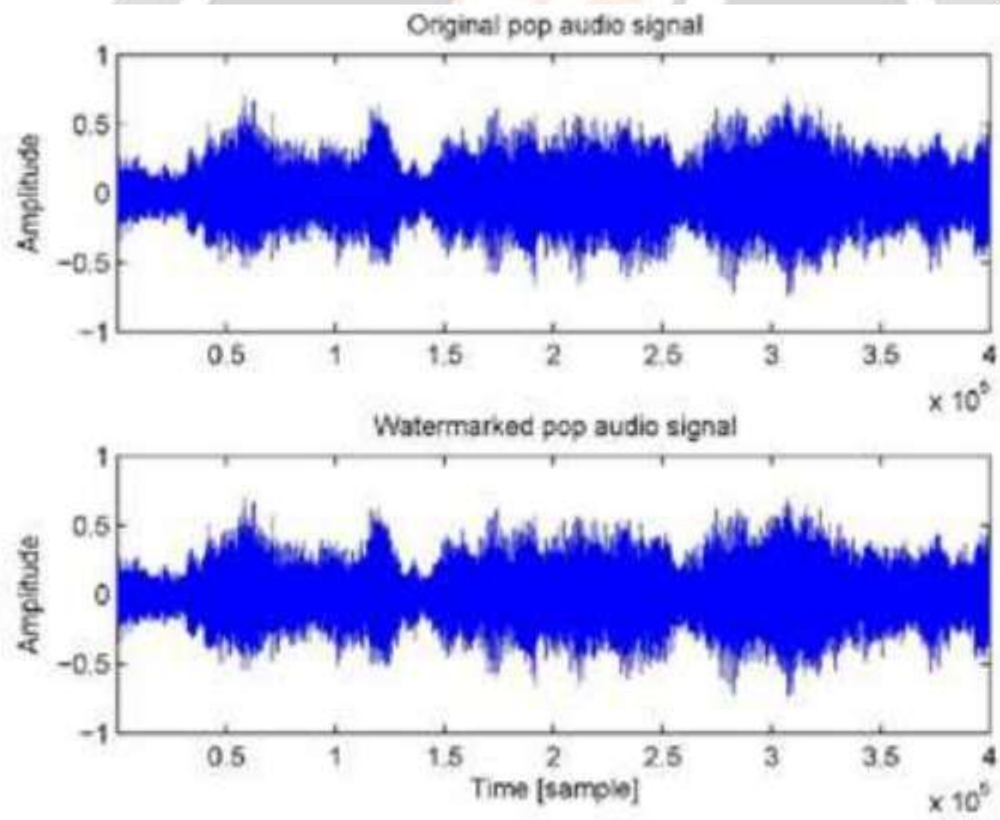
To evaluate the water-mark detection accuracy after attacks, we used the and the de-fined as follows [4]: The original watermark and the extracted one we use the measure defined as follows:

Step 1: Split the watermarked signal into frames.

Step 2: Decompose each frame into IMFs.

Step 3: Extract the extrema of audio.

indicates the presence of watermark while a low value suggests the lack of watermark. Two types of errors may occur while searching the SCs: the False Positive Error (FPE) and the False Neg-ative Error (FNE). These errors are very harmful because they impair the credibility of the watermarking system. The associated probabili-ties of these errors are given by [3], [4]: where is the SC length and is is the threshold. is the proba-bility that a SC is detected in false location while is the proba-bility that a watermarked signal is declared as unwatermarked by the decoder. We also use as performance measure the payload which quan-tifies the amount of information to be hidden. More precisely, the data payload refers to the number of bits that are embedded into that audio signal within a unit of time and is measured in unit of bits per second (b/s).



6. RESULTS

The retrieved audio signal is representation of audio frequency range of roughly 20 to 20,000 Hz. Audio signals may be synthesized directly, or may originate at a transducer. The signal flow is the path an audio signal will take from source or recording devices. Audio signals may be characterized by parameters such as their bandwidth, power level in decibels and voltage level. For watermark extraction, host signal is spitted into frames and EMD is performed on each one as in embedding. We then search for SCs in the extracted data. This procedure is repeated by shifting the selected segment (window) one sample at time until a SC is found. With the position of SC determined, we can then extract the hidden information bits, which follow the synchronization code. Split the watermarked signal into frames. Decompose each frame into IMFs. Extract the extreme { } * i e of IMFc. Set the start index of the extracted data, y, to I=1 and select N1=L samples (sliding window size). Extract the P watermarks and make comparison bit by bit between these marks, for correction, and finally extract the desired watermark.

7. CONCLUSION

In this paper audio watermarking using EMD and DCT is proposed. The data driven decomposition property of EMD and energy compaction property of DCT is utilized. This method offers consistent performance for all types of audio as the EMD decomposition is not pre deterministic and only based on the data values. The performance is evaluated based on imperceptibility, robustness and payload. The proposed method significantly improved the payload in comparison to that obtained by some existing method. The watermark is robust under attacks such as AWGN, MP3 compression. As the SNR is significantly high between the watermarked and original audio, the audio is not audible to the listener.

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